List of Claims:

Claim 1 (currently amended): A method of coding a speech signal, the method comprising the steps of:

accumulating samples of the speech signal over a sampling duration to provide accumulated samples;

evaluating the accumulated samples to obtain a representative sample;

determining whether a slope of the representative sample conforms to a defined characteristic slope stored in a reference database of spectral characteristics; and

selecting a value of a coding parameter, for coding the speech signal, based on the determining step;

wherein the selecting step selects a first coding parameter value as the value if the determining step determines that the slope of the representative sample of the speech signal conforms to the defined characteristic slope, and wherein the selecting step selects a second coding parameter value as the value if the determining step determines that the slope of the representative sample of the speech signal is generally flat.

Claims 2-3 (cancelled)

Claim 4 (previously presented): The method according to claim 1 where the evaluating comprises averaging the accumulated samples over the sampling duration to obtain the representative sample.

Claim 5 (original): The method according to claim 1 further comprising the step of assuming the spectral response of a speech signal is sloped in accordance with the defined

characteristic slope prior to completion of at least one of the accumulating step and the determining step.

Claim 6 (currently amended): The method according to claim 1 wherein the selecting step comprises selecting ~~a~~ the first coding parameter value as the value of an initial default coding parameter based on the assumption that the spectral response of the speech signal is sloped in accordance with the defined characteristic slope.

Claim 7 (currently amended): The method according to claim 1 where the defined characteristic slope approximately represents a Modified Intermediate ~~Response~~ Reference System.

Claim 8 (original): The method according to claim 1 wherein the selecting comprises selecting at least one preferential encoding parameter value as the value; an encoding parameter underlying the at least one preferential encoding parameter value and including one or more of the following: pitch gain per frame or subframe, at least one filter coefficient of a perceptual weighting filter, at least one bandwidth expansion constant associated with a synthesis filter, and at least one bandwidth expansion constant associated with an analysis filter.

Claim 9 (original): The method according to claim 1 where the selecting comprises selecting at least one preferential decoding parameter value as the value; a decoding parameter underlying at least one decoding parameter value and including one or more of the following: at least one bandwidth expansion constant associated with a synthesis filter and at least one linear predictive filter coefficient associated with a post filter.

Claim 10 (currently amended): The method according to claim 1 where the selecting comprises adjusting the value of ~~a~~ the coding parameter selected from the group consisting of

pitch gains per frame or subframe, at least one filter coefficient of a perceptual weighting filter, at least one bandwidth expansion constant associated with a synthesis filter, at least one bandwidth expansion constant associated with an analysis filter, and at least one linear predictive filter coefficient associated with a post filter.

Claim 11 (original): The method according to claim 1 further comprising adjusting a bandwidth expansion of the speech signal as the value for at least one of a synthesis filter and an analysis filter from a previous value to a revised value based on a degree of slope or flatness in the speech signal.

Claim 12 (original): The method according to claim 1 where the selecting comprises selecting a bandwidth expansion value of the speech signal as the value in conformance with the following equations:

$$\frac{1}{A(z)} = \frac{1}{1 - \sum_{i=1}^{P} a_{i\ \mathrm{revised}} z^{-i}},$$

where 1/A(z) is a filter response represented by a z transfer function, ai revised is a linear predictive coefficient, $i = 1...P$, and P is the prediction order or filter order of the synthesis filter,

$$a_{i\ \mathrm{revised}} = a_{i\ \mathrm{previous}} \gamma^{i},$$

where $a_{i\ \mathrm{revised}}$ is a revised linear predictive coefficient, $a_{i\ \mathrm{previous}}$ is a previous linear predictive coefficient, γ is the bandwidth expansion constant, $i = 1...P$, and P is the prediction order of the synthesis filter of the encoder, and where $a_{i\ \mathrm{previous}}$ represents a member of the set of extracted linear predictive coefficients $\{a_{i\ \mathrm{previous}}\}^{P}_{i=1}$, for the synthesis filter of the encoder.

Claim 13 (original): The method according to claim 12 where the value of the bandwidth expansion constant for a generally flat spectral response differs from that of the defined characteristic slope.

Claim 14 (original): The method according to claim 12 where the value of the bandwidth expansion constant is greater for a generally flat spectral response than the defined characteristic slope.

Claim 15 (original): The method according to claim 12 where γ is set to a first value of approximately .99 if the slope of the representative sample is consistent with an MIRS spectral response and γ is set to a second value of approximately .995 where the slope of the representative sample is generally flat or approaches zero.

Claim 16 (original): The method according to claim 1 wherein the selecting comprises selecting a frequency response factor of a perceptual weighting filter as the value of the coding parameter based on a degree of slope or flatness in the speech signal.

Claim 17 (original): The method according to claim 1 further comprising controlling a frequency response of a perceptual weighting filter based on the following equation:

$$W(z) = \frac{1}{1-\alpha z^{-1}} \frac{1+\sum_{i=1}^{P} a_i \rho^i z^{-i}}{1+\sum_{i=1}^{P} a_i \beta^i z^{-i}}$$

where α is a weighting constant as the value of the coding parameter, β and ϱ are preset coefficients, P is the predictive order, and $\{a_i\}$ is the linear predictive coding coefficient.

Claim 18 (original): The method according to claim 17 wherein the controlling comprises selecting different values of the weighting constant α to adjust the frequency response

of the perceptual weighting filter in response to the determined slope or flatness of the speech signal.

Claim 19 (original): The method according to claim 17 further comprising controlling the value of α based on the spectral response of the speech signal such that α approximately equals .2 where the speech signal is consistent with the MIRS spectral response and α approximately equals 0 where the speech signal is consistent with a generally flat signal response.

Claim 20 (original): The method according to claim 1 further comprising the step of selecting a frequency response factor of a post filter as the value of the coding parameter based on a degree of slope or flatness of the speech signal.

Claim 21 (original): The method according to claim 1 further comprising the step of controlling a frequency response of a post filter in accordance with the following equation:

$$P(z) = \frac{1 + \sum_{i=1}^{P} a_i \gamma_1^i z^{-i}}{1 + \sum_{i=1}^{P} a_i \gamma_2^i z^{-i}}$$

where γ_1 and γ_2 represents a set of post-filtering weighting constants in which the value is a member of the set, {ai} is the linear predictive coding coefficient, and P is the filter order of the post filter.

Claim 22 (original): The method according to claim 21 further comprising the step of controlling a frequency response of a post filter by selecting different values of post-filtering weighting constants of γ_1 and γ_2 in response to the determined slope or flatness of the speech signal.

Claim 23 (original): The method according to claim 21 where γ_1 and γ_2 approximately equal .65 and .4, respectively, if the speech signal is consistent with an MIRS spectral response; and where γ_1 and γ_2 approximately equal .63 and .4, respectively, if the speech signal is consistent with a generally flat signal response.

Claim 24 (currently amended): A system for coding a speech signal, the system comprising:

a buffer memory for accumulating samples of the speech signal over a sampling duration to provide accumulated samples;

an evaluator adapted to evaluate the accumulated samples to obtain a representative sample and to make a determination whether a slope of the representative sample of the speech signal conforms to a defined characteristic slope stored in the storage device; and

a selector for selecting a preferential one of a first coding parameter value and a second coding parameter value for coding the speech signal based on the determination;

wherein the selector selects a first coding parameter value as the value if the evaluator determines that the slope of the representative sample of the speech signal conforms to the defined characteristic slope, and wherein the selector selects a first coding parameter value as the value if the evaluator determines that the slope of the representative sample of the speech signal is generally flat.

Claims 25-26 (cancelled)

Claim 27 (previously presented): The system according to claim 24 where the evaluator comprises an averaging unit adapted to average the accumulated samples over the sampling duration to obtain the representative sample.

Claim 28 (previously presented): The system according to claim 24 where the evaluator assumes the spectral response of a speech signal is sloped in accordance with the defined characteristic slope prior to the expiration of the sampling duration.

Claim 29 (currently amended): The system according to claim 24 where the defined characteristic slope approximately represents a Modified Intermediate ~~Response~~ Reference System.

Claim 30 (original): The system according to claim 24 where the evaluator triggers an adjustment of at least one encoding parameter to a revised encoding parameter during the coding process.

Claim 31 (original): The system according to claim 24 where the evaluator is coupled to a coder, where the evaluator sends at least one of a control data and a spectral-content indicator to the coder for controlling one or more of the following coding parameters: (a) pitch gains per frame or subframe, (b) at least one filter coefficient of a perceptual weighting filter of an encoder, (c) at least one filter coefficient of a synthesis filter of an encoder, (d) at least one bandwidth expansion constant associated with a synthesis filter of the coder, (e) at least one bandwidth expansion constant associated with a synthesis filter of a decoder, (f) at least one bandwidth expansion constant associated with an analysis filter of an encoder, and (g) at least one filtering coefficient associated with a post filter coupled to a decoder.